LISTING OF THE CLAIMS

This listing of claims will replace all prior versions, and listings, of claims in the application:

1-10. (Cancelled)

11. (Currently Amended) A microphone array processing system for performance enhancement in noisy environments, the system comprising:

a plurality of microphones positioned to detect speech from a single speech source and noise from a noise source, and to generate corresponding microphone output signals, one of the microphones being designated a reference microphone and the others being designated data microphones, wherein the reference microphone and the data microphones receive acoustic signals both from the speech source and from the noise source;

a plurality of bandpass filters, one for each microphone, for eliminating from the microphone output signals a known spectral band containing noise;

a plurality of adaptive filters, one for each of the data microphones, for aligning each data microphone output signal with the output signal from the reference microphone; and

a signal summation circuit for combining the adaptively filtered output signals from the microphones, ~~whereby~~such that signal components resulting from the speech source combine coherently and signal components resulting from noise combine incoherently, to produce an increased signal-to-noise ratio.

12. (Previously Presented) A system as defined in claim 11, and further comprising speech detection circuitry for enabling the plurality of adaptive filters only when speech is detected.

13. (Previously Presented) A system as defined in claim 11, and further comprising speech conditioning circuitry coupled to the signal summation circuit to reduce reverberation effects in

the output signal by modifying the spectrum of the cumulative signal obtained from the signal summation circuit.

14. (Previously Presented) A system as defined in claim 13, wherein each of the adaptive filters comprises:

means for filtering data microphone output signals by convolution with a vector of weight values in the frequency domain;

means for comparing the filtered data microphone output signal from one of the data microphones with an output signal from the reference microphone in the frequency domain-and deriving therefrom an error signal; and

means for adjusting the weight values convolved with the data microphone output signals in the frequency domain to minimize the error signal.

15. (Previously Presented) A system as defined in claim 14, wherein each of the adaptive filters further includes Fast Fourier Transform means to transform successive blocks of data microphone output signals to a frequency domain representation to facilitate filtering in the frequency domain.

16. (Currently Amended) A method for improving detection of speech signals, the method comprising:

positioning a plurality of microphones to detect speech from a single speech source and noise from a noise source, one of the microphones being designated a reference microphone and the others being designated data microphones, wherein the reference microphone and the data microphones receive acoustic signals both from the speech source and from the noise source;

generating microphone output signals in the microphone;

filtering the microphone output signals in a plurality of bandpass filters, one for each microphone, to eliminate from the microphone output signals a known spectral band containing noise;

adaptively filtering the microphone output signals in a plurality of adaptive filters, one for each of the data microphones, and thereby aligning each data microphone output signal with the output signal from the reference microphone; and

combining the adaptively filtered output signals from the microphones in a signal summation circuit, ~~whereby~~such that signal components resulting from the speech source combine coherently and signal components resulting from noise combine incoherently, to produce an increased signal-to-noise ratio.

17. (Previously Presented) A method as defined in claim 16, further comprising:

detecting speech received by the microphones; and

enabling the step of adaptively filtering the microphone signals only when speech is detected.

18. (Previously Presented) A method as defined in claim 16, and further comprising conditioning the combined signals in speech conditioning circuitry coupled to the signal summation circuit, to reduce reverberation effects in the output signal by modifying the spectrum of the cumulative signal obtained from the signal summation circuit.

19. (Previously Presented) A method as defined in claim 18, wherein the adaptive filtering further comprises:

filtering data microphone output signals by convolution with a vector of weight values in the frequency domain;

comparing the filtered data microphone output signals from one of the data microphones with reference microphone output signals in the frequency domain and deriving therefrom an error signal;

adjusting the weight values convolved with the data microphone output signals to minimize the error signal; and

repeating the filtering, comparing and adjusting steps to converge on a set of weight values that results in minimization of noise effects.

20. (Previously Presented) The method as defined in claim 19, wherein the adaptive filtering further comprises:

obtaining a block of data microphone signals;

transforming the block of data to a frequency domain using a Fast Fourier Transform;

filtering the block of data in the frequency domain using a current best estimate of weighting values;

comparing the filtered block of data with corresponding data derived from the reference microphone in the frequency domain;

updating the filter weight values to minimize any difference detected in the comparing step in the frequency domain;

transforming the filter weight values back to the time domain using an inverse Fast Fourier transform;

zeroing out portions of the filter weight values that give rise to unwanted circular convolution; and

converting the filter values back to the frequency domain.

21. (Previously Presented) A method for improving detection of speech signals, the method comprising:

converting a reference microphone data to a frequency domain;

converting data microphone data to the frequency domain.

filtering the data microphone data with a current filter weight value in the frequency domain;

updating the filter weight value with the reference microphone data;

converting the updated filter weight value from the frequency domain to a time domain;

zeroing out portions of the weight value; and

converting the weight value back to the frequency domain.

22. (Previously Presented) The adaptive filter process of claim 21, the converting a reference microphone data to a frequency domain comprises transforming the reference microphone data via a Fast Fourier transform.

23. (Previously Presented) The adaptive filter process of claim 21, the converting data microphone data comprises transforming the reference microphone data via a Fast Fourier transform.

24. (Previously Presented) The adaptive filter process of claim 21, the updating the filter weight value comprises $W(k+1) = W(k)+\mu(Ref(k) - X(k))*conj(Y)$, where k is the data block number and μ is a small adaptive step constant.

25. (Previously Presented) The adaptive filter of claim 21, the converting the filter weight value from the frequency domain to the time domain comprises transforming via an inverse Fast Fourier transform.

26. (Previously Presented) The adaptive filter process of claim 21, wherein the zeroing out portions of the weight value zeroes out portions that give rise to circular convolution.

27. (Previously Presented) The adaptive filter process of claim 21, wherein the updating the filter weight value with the reference microphone data comprises

comparing the filtered data microphone data with reference microphone data in the frequency domain and deriving therefrom an error signal; and

adjusting the weight values convolved with the data microphone output signals to minimize the error signal.

28. (Previously Presented) A system for detecting speech signals comprising:

means for converting reference microphone data from a reference microphone to a frequency domain;

means for converting data microphone data from a data microphone to the frequency domain.

means for filtering the block of data from the data microphone with a current filter weight value in the frequency domain;

means for updating the filter weight value with the reference microphone data in the frequency domain;

means for converting the updated filter weight value from the frequency domain to the time domain;

means for zeroing out portions of the weight value in the time domain; and

means for converting the weight value back to the frequency domain.

29. (Previously Presented) The adaptive filter of claim 28, means for converting reference microphone data from a reference microphone to a frequency domain further comprising a Fast Fourier transform means.

30. (Previously Presented) The adaptive filter of claim 28, the means for converting data microphone data from a data microphone to the frequency domain comprises a Fast Fourier transform means.

31. (Previously Presented) The adaptive filter of claim 28, the updating the filter weight value comprises $W(k+1) = W(k)+\mu(Ref(k) - X(k))*conj(Y)$, where k is the data block number and μ is a small adaptive step constant.

32. (Previously Presented) The adaptive filter of claim 28, means for converting the filter weight value from the frequency domain to a time domain comprises an inverse Fast Fourier transform means.

33. (Previously Presented) The adaptive filter of claim 28, wherein the updating the filter weight value with the reference microphone data comprises

means for comparing the filtered data microphone data with reference microphone data in the frequency domain and deriving therefrom an error signal; and

means for adjusting the weight values convolved with the data microphone output signals to minimize the error signal.